Configuration Notes 245

Mediatrix 4400 Digital Gateway
VoIP Gateway with the PSTN

January 16, 2007

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Introduction

This document outlines the configuration steps to set up a Mediatrix® 4400 digital gateway to act as a gateway with the PSTN.

Mediatrix 4400 Digital Gateway Overview

These configuration notes apply to the Mediatrix 4400 Series digital gateway products. The Mediatrix 4400 Series Digital Gateways allow enterprises to lower communications costs over any IP link. The platform features ISDN BRI interfaces. They provide an ideal solution for enterprise voice applications or for connecting to a service provider's broadband access.

Mediatrix® 4400 digital gateways are fully scalable in terms of number of ports and functionalities. They currently come in the following models:

Model	Interfaces	VoIP Call Capacity
Mediatrix 4401	1 BRI port	up to 2
Mediatrix 4402	2 BRI ports	up to 4
Mediatrix 4404	4 BRI ports	up to 8

The Mediatrix digital gateways link any standard ISDN BRI connection to the IP network and deliver the clarity of toll quality voice for a comprehensive VoIP solution.

T.38 FoIP, fax bypass, and modem bypass capabilities ensure that the Mediatrix digital gateways seamlessly transport voice and data services. The Mediatrix digital gateways offer flexibility and scalability for VoIP network integration and low bandwidth voice.

With configurable NT/TE BRI ports, call-switching, and user programmable call routing (including caller/called ID), Mediatrix digital gateways integrate smoothly into existing PBX and PSTN networks.

Key Features:

- Voice Routing
- Fax over IP support, including T.38
- Proven voice algorithms implemented on dedicated DSP for enhanced voice quality
- Up to 8 simultaneous calls
- SNMPv3 and web management
- Configuration file encryption
- Automatic firmware and configuration file download
- Optional PSTN Bypass feature
- Optional Power Over Ethernet
- Optional Power Feeding Module for BRI phones

Deployment Scenario

Description

This configuration note is a step-by-step guide to set up one Mediatrix 4402 digital gateway to act as a gateway with the PSTN. The Mediatrix 4402 is used to connect an existing VoIP network to the PSTN. The configuration starts with the Mediatrix 4402 default configuration but can be easily customized for the 4404 and 4401, so from now on, the device will be referred to as the *Mediatrix 4400*. The following is the network topology to which we will refer in our sample deployment.

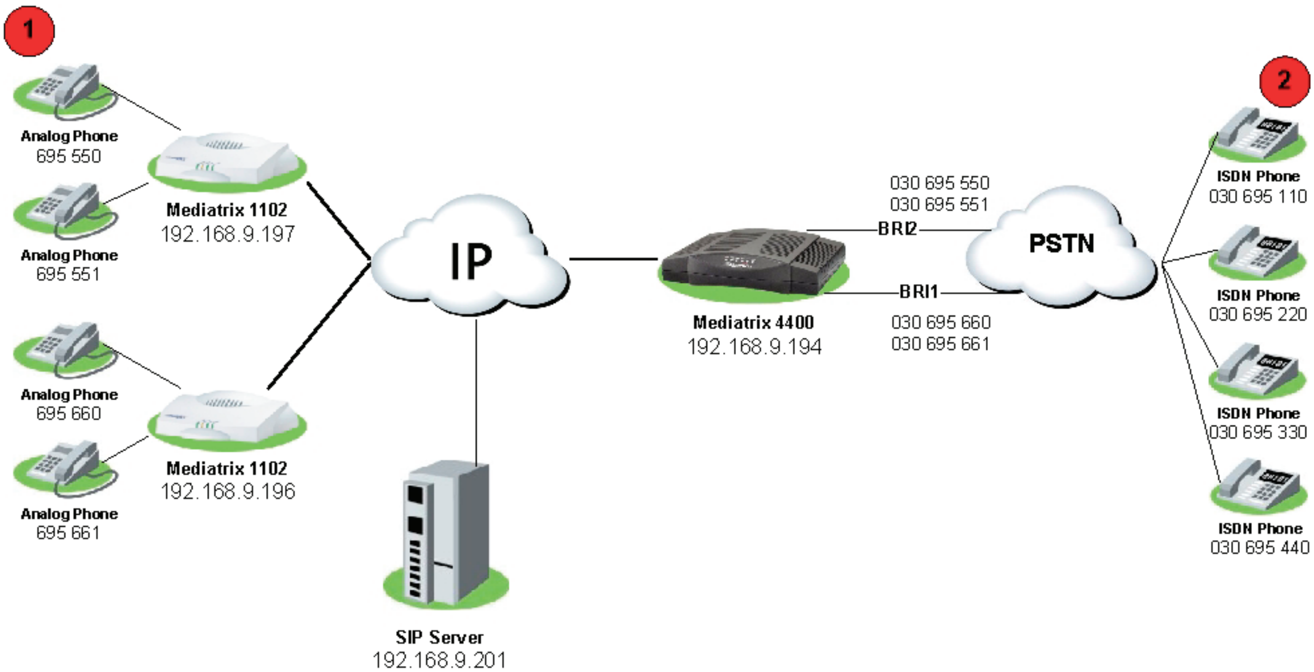


Figure 1 - Network Topology

Note: The network addresses and phone numbers shown above are sample values that will most probably vary in your specific setup. In the following pages, when referring to such a sample value, it will be visually outlined (e.g., ***192.168.9.194***), so whenever you see parameters outlined in that fashion, you should replace them with the values that are appropriate for your specific setup.

Objectives

The steps described in the following pages will show you how to setup the Mediatrix 4400 so it can:

A. receive calls from the PSTN and route them to the VoIP network (e.g., phone 2 calls phone 1):
   1. a user picks up an ISDN phone and dials the number of a line connected to the Mediatrix 4400.
   2. the PSTN routes the call to the Mediatrix 4400.
   3. the Mediatrix 4400 routes the call to the appropriate Mediatrix 1102, removing necessary prefix (area code ***030***) from the called number.
   4. the Mediatrix 1102 makes the appropriate analog phone ring.
   5. a user picks up the analog phone and the call is established.

B. receive calls from the VoIP network and route them to the PSTN on one of the BRI lines connected to the Mediatrix 4400 (e.g. phone 1 calls phone 2):
   1. a user picks up an analog phone and dials a number.
   2. the appropriate Mediatrix 1102 routes the call to the Mediatrix 4400.

3. the Mediatrix 4400 decides to which ISDN BRI interface route this call.
4. the Mediatrix 4400 adds the necessary prefix (area code ***030***) to the dialled phone number before routing the call to the PSTN.
5. the PSTN makes the appropriate ISDN phone ring.
6. a user picks up the ISDN phone and the call is established.

Assumptions

This configuration note focuses on configuring the Mediatrix 4400 and assumes that:

- the SIP server's default route's destination is the Mediatrix 4400, e.g., when a number dialed on an analog phone is unknown, the call is sent to the Mediatrix 4400.
- the left-hand side of the VoIP network setup is functional (Mediatrix 1102, analog phones, SIP server), and the SIP users are correctly registered to the SIP server.

Steps

This configuration note will guide you through the following steps:

1. Physical connection of the Mediatrix 4400 to the network and PBX.
2. IP address discovery or configuration.
3. Web interface access.
4. SIP configuration.
5. ISDN configuration.
6. Call routing configuration.
7. Basic call establishment.

Configuration of the Mediatrix 4400 Digital Gateway

Physical Connection of the Mediatrix 4400 to the Network and PSTN

Please refer to the Mediatrix 4400 Quick Start booklet (packaged with the Mediatrix 4400) for instructions on hardware installation.

The Mediatrix 4400 Quick Start booklet can also be found online on the Mediatrix Download Portal at https://support.mediatrix.com/DownloadPlus/Download.asp.

IP Address Discovery or Configuration

The purpose of this section is to be able to contact the Mediatrix 4400's management interface to start with unit configuration.

Once the physical connection is complete and the Mediatrix 4400 is powered up, the first thing to do is find out the IP address the Mediatrix 4400 is using. The Mediatrix 4400's IP address can be set either dynamically or statically. The default behaviour of the Mediatrix 4400 is to try to obtain a dynamic IP address through DHCP.

Dynamic IP Address Discovery

Before connecting the Mediatrix 4400 to the network, Mediatrix strongly suggests that you reserve an IP address in your DHCP server for the unit you are about to connect. DHCP servers reserve IP addresses for specific devices by using a unique identifier for each device. The Mediatrix 4400's unique identifier is its media access control (MAC) address. The MAC address appears on the label located on the bottom side of the unit.

If you have not reserved an IP address, you can discover which IP address has been assigned to the Mediatrix 4400 by either:

- consulting your DHCP server's logs to find out details on the DHCP lease that was given to the Mediatrix 4400.
- using a network packet sniffer (e.g., Ethereal) to examine the DHCP messages exchanged between the Mediatrix 4400 and your DHCP server while the Mediatrix 4400 boots up.

Default Static IP Address Configuration

If there is no DHCP server in your network, then the IP address has to be configured statically. The first thing to do is set the Mediatrix 4400 to its known default static IP address. You can do this by using the Mediatrix 4400's partial reset feature (see the section Further Information and Configuration for more details).

1. Once the Mediatrix 4400 has finished booting up (the *Power* LED is lit, not blinking), insert a small, unbent paper clip into the RESET/DEFAULT hole located at the rear of the Mediatrix 4400 to press the RESET/DEFAULT button. The *Power* LED will start blinking, and after a few seconds, all the LEDS will start blinking. Release the paper clip after all the LEDs start blinking and before they all stop blinking (between 7-11 seconds).

After a partial reset is performed, the Mediatrix 4400 uses the default IP address 192.168.0.1. From now on, you can optionally change the Mediatrix 4400's IP address (see section Further Information and Configuration for more details).

Web Interface Access

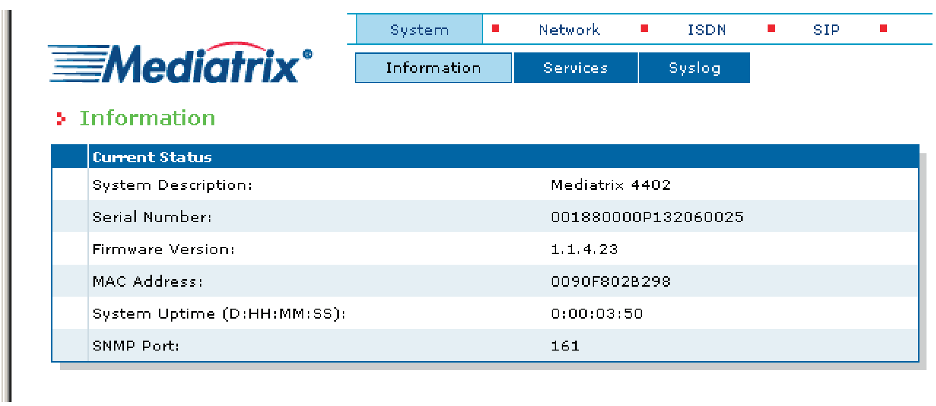
The purpose of this section is to log in to the Mediatrix 4400's web interface.

The Mediatrix 4400's web interface can be used to view the status of the Mediatrix 4400 and set its numerous parameters.

1. In your web browser's address field, type ***192.168.9.194*** (or the address of the Mediatrix 4400). The PC you use must be connected to the same subnet as the Mediatrix 4400 or to a network where it can reach the Mediatrix 4400's IP address. The following window appears:

2. Enter the user name **public**. Leave the *Password* field empty.

3. Click **Login**. The following window appears.



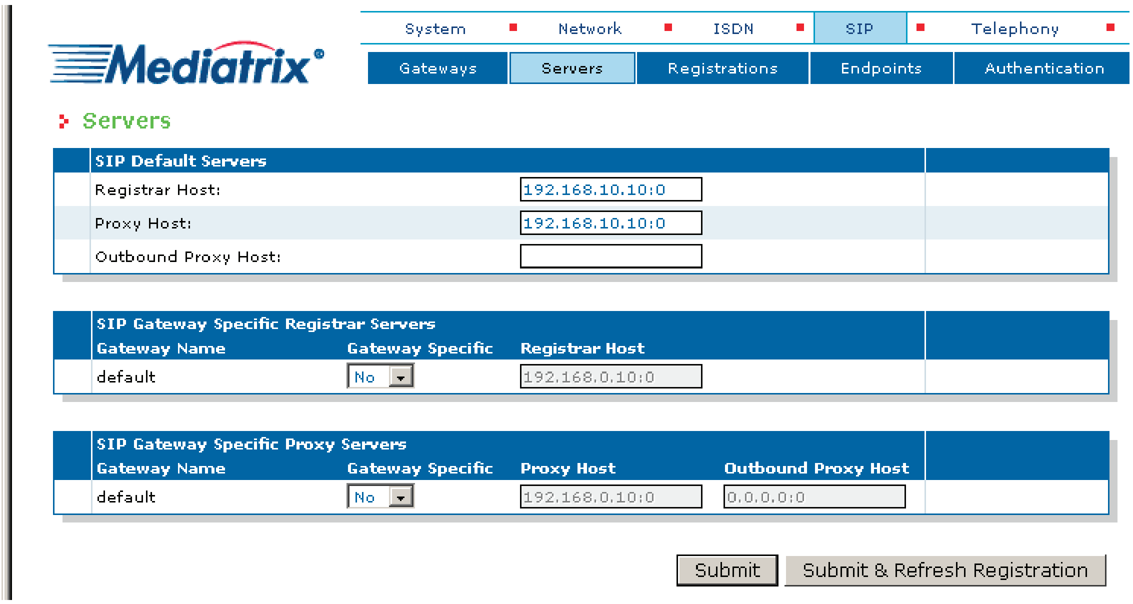
You now have access to the Mediatrix 4400's configuration web interface.

SIP Configuration

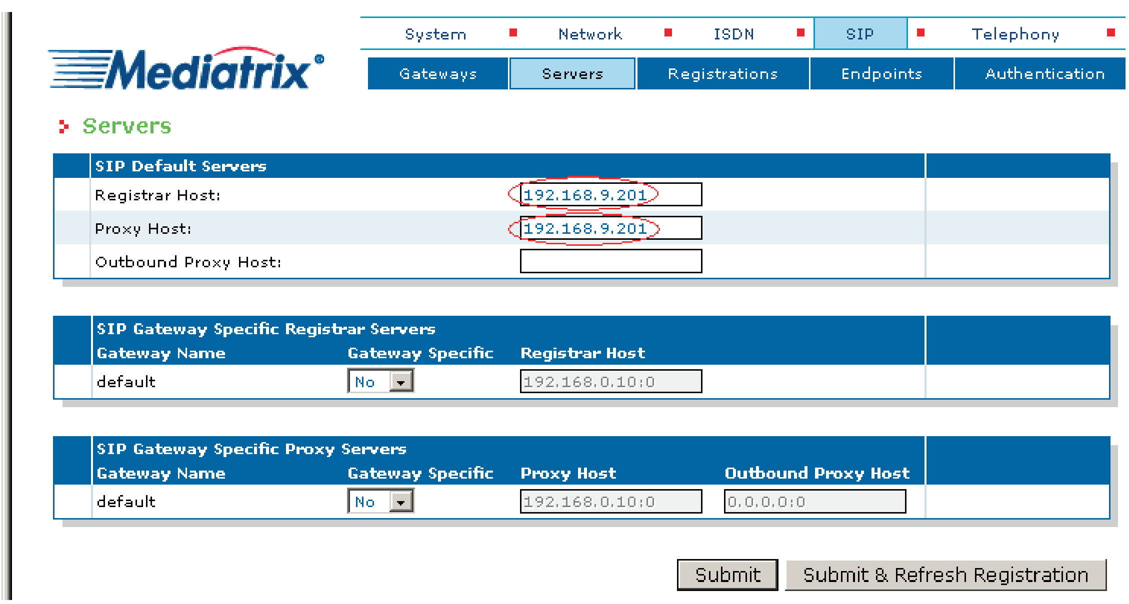
The purpose of this section is to setup the Mediatrix 4400 to use your SIP server for registration and call routing.

The SIP configuration tells the Mediatrix 4400 which SIP servers, parameters and phone numbers to use. The following steps configure the Mediatrix 4400 as illustrated in the sample network topology.

1. Click the **SIP** menu, then the **Servers** sub-menu. The following window appears:



2. Set the *Registrar Host* field to the address of the central SIP Server ***192.168.9.201***.
3. Set the *Proxy Host* field to the address of the central SIP Server ***192.168.9.201***.



4. Click **Submit** to save the configuration changes. The Mediatrix 4400 is now configured to use your SIP server.

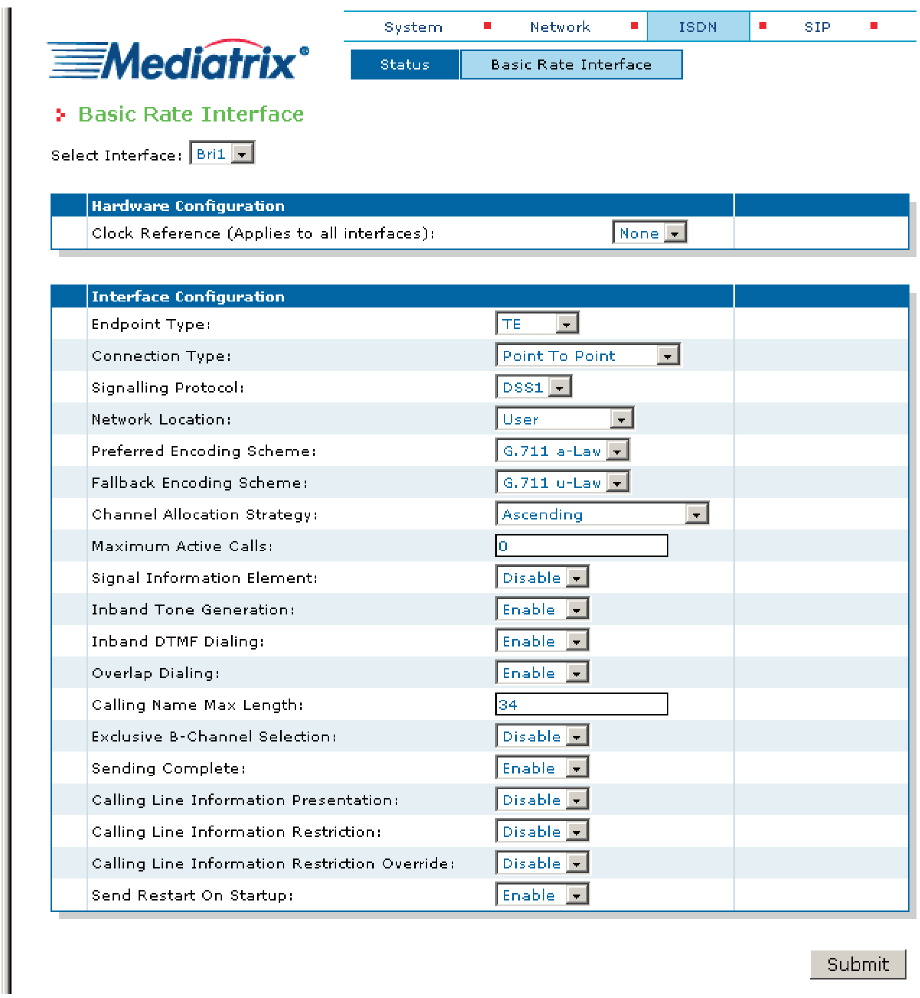
5. OPTIONAL STEP: if your SIP server requires SIP authentication, further configuration steps are necessary so the Mediatrix 4400 has all the needed information to authenticate to the server (see the section Further Information and Configuration for more details).

ISDN Configuration

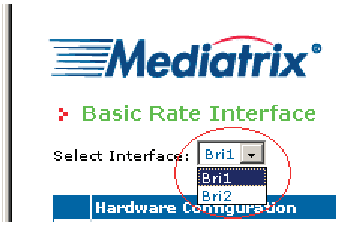
The purpose of this section is to configure the Mediatrix 4400's ISDN BRI interfaces in Terminal mode (TE) for a point-to-point line. The PSTN line to which the Mediatrix 4400 is connected must be a point- to-point line in Network mode (NT). If your setup differs, please refer to the section Further Information and Configuration for more details.

The ISDN configuration tells the Mediatrix 4400 how its ISDN BRI interfaces should behave. You must configure the ISDN parameters of the Mediatrix 4400 digital gateways for each interface you intend to use.

1. Click the **ISDN** menu, then the **Basic Rate Interface** sub-menu. The following window appears:

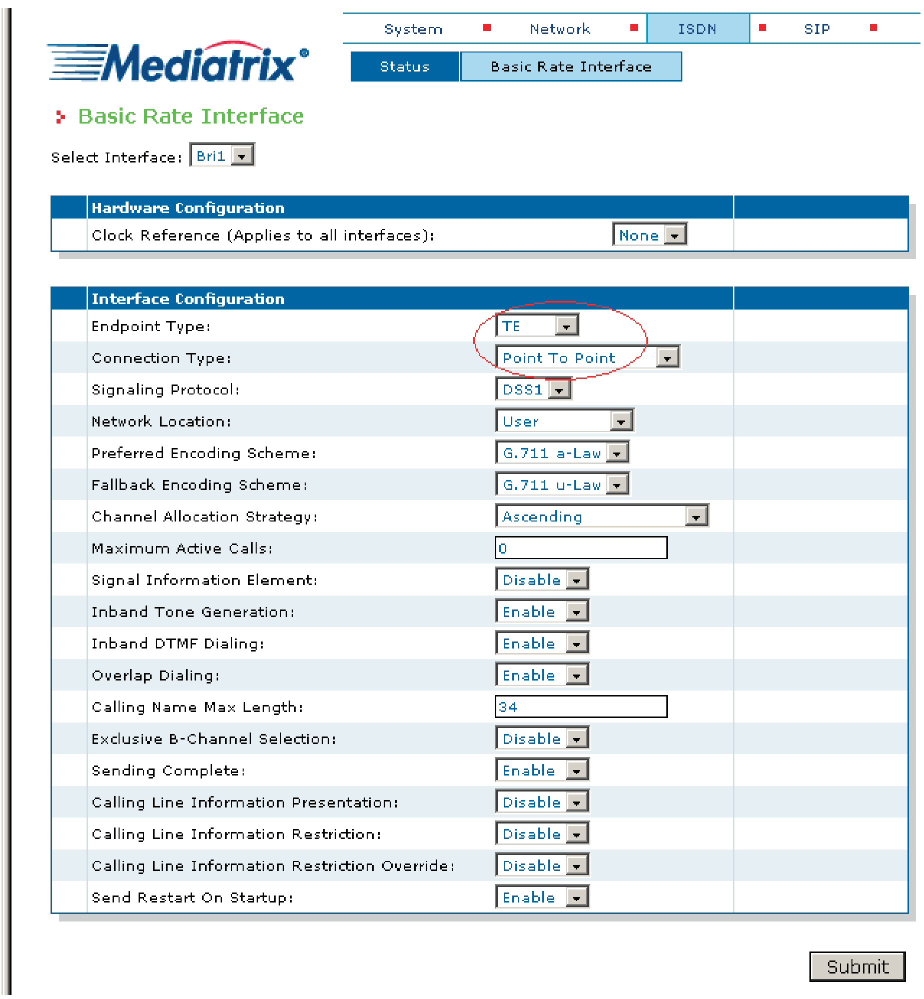


2. Select the interface for which you want to apply the changes in the *Select Interface* drop-down menu. Depending on the model of Mediatrix 4400 you are using, you may have 1, 2, or 4 interfaces available in the drop-down menu.

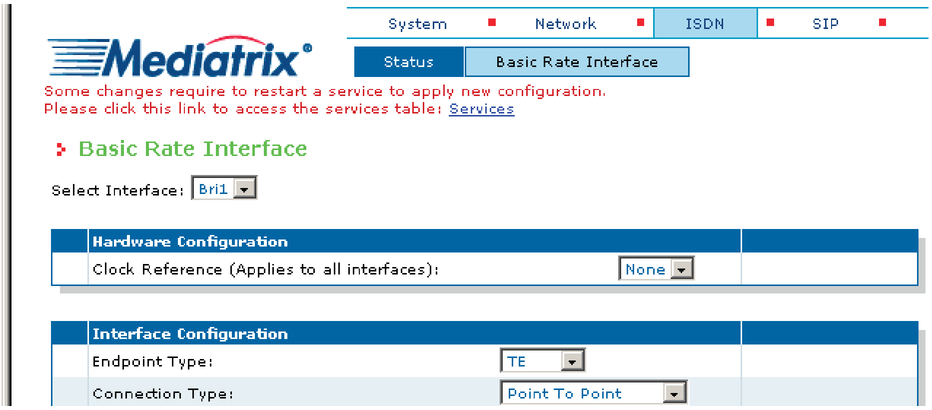


3. In the *Interface Configuration* section, set the *Endpoint Type* field to **TE** and *Connection Type* field to **Point to Point**. Leave all other parameters to their default values.

> **NOTE**: Depending on the model of Mediatrix 4400 you are using, there may or may not be a *Power Feeding* field in the *Hardware Configuration* section.



4. Click **Submit** to apply the configuration changes made to this interface.



5. The parameters that have just been configured require a restart of the ISDN service. A service is a logical group of features. Restarting a service is a required mechanism for certain elements in the configuration. However, you can finish with the ISDN configuration steps before doing that. Once the ISDN configuration is ready, follow the instructions from Appendix A - Restarting a Service to restart the ISDN service as required.
6. Repeat steps 2 to 4 for all of the ISDN BRI interfaces listed in the *Select Interface* field.
7. Restart the ISDN service as described in Appendix A - Restarting a Service.

Call Routing Configuration

The purpose of this section is to configure the Mediatrix 4400's call router so it can route calls to/from the VoIP network and the PSTN as described in the Deployment Scenario section.

You must configure the call router parameters of the Mediatrix 4400 digital gateways so that the calls can properly terminate. Remember that the purpose of this configuration note is to achieve the sample deployment scenario shown in Figure 1. Your specific setup may vary.

Planning the Call Router

The goal of planning the Call router configuration is to summarize the rules incoming calls will follow when passing through the Mediatrix 4400.

This is:

- Call sources and destinations.
- Calls allowed and rejected.
- Call properties manipulations.
- All routing possibilities.

Before going further with the configuration steps, you should refer back to the two types of calls described in the Deployment Scenario section.

The most basic call scenario implies at least the configuration of *Routes*. In the current deployment scenario, you will also configure *Mappings* to support Step 3 of call scenario A, and a *Hunt Group* to support Step 3 of call scenario B defined in the Deployment Scenario section (see Further Information and Configuration for more details).

- A Route is a virtual connection made inside the Mediatrix 4400 between call sources and destinations. Routes are part of the Mediatrix 4400's Route table. When a call comes in, the Mediatrix 4400 uses its Route table to decide to which destination the call must be forwarded.
- A Hunt Group is a virtual entity that regroups different call destinations into one group. This entity can then be used as a call destination in a Route. When an incoming call is routed to a Hunt, the Hunt group selects one of its available destinations to route the call.

- A Mapping is a transformation that can be applied to a call when it goes through one of the Mediatrix 4400's routes, according to various criteria. It can be used, for example, to apply changes to the call's dialed phone number.

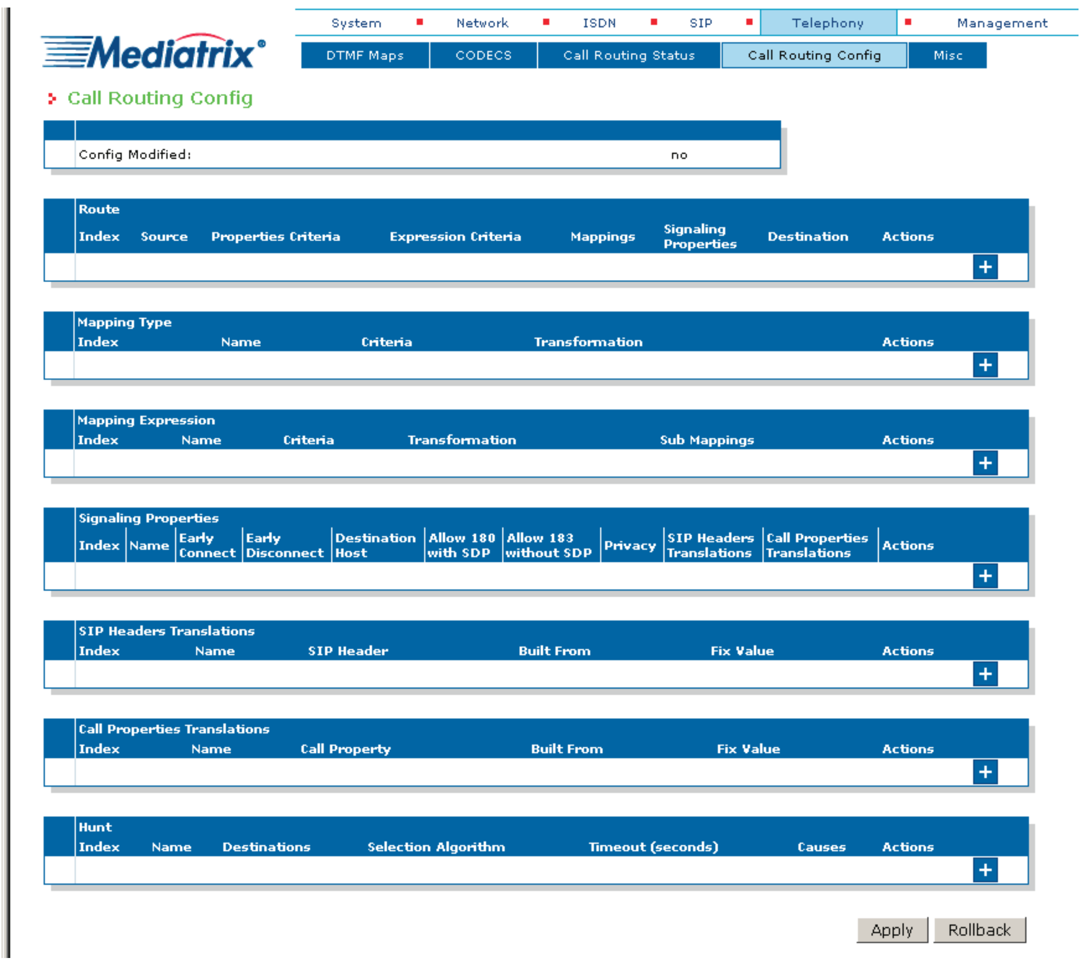
Configuring the Call Router

Hunt Group

The purpose of this subsection is to configure a Hunt Group in the Mediatrix 4400, so it can be later used as a route's destination.

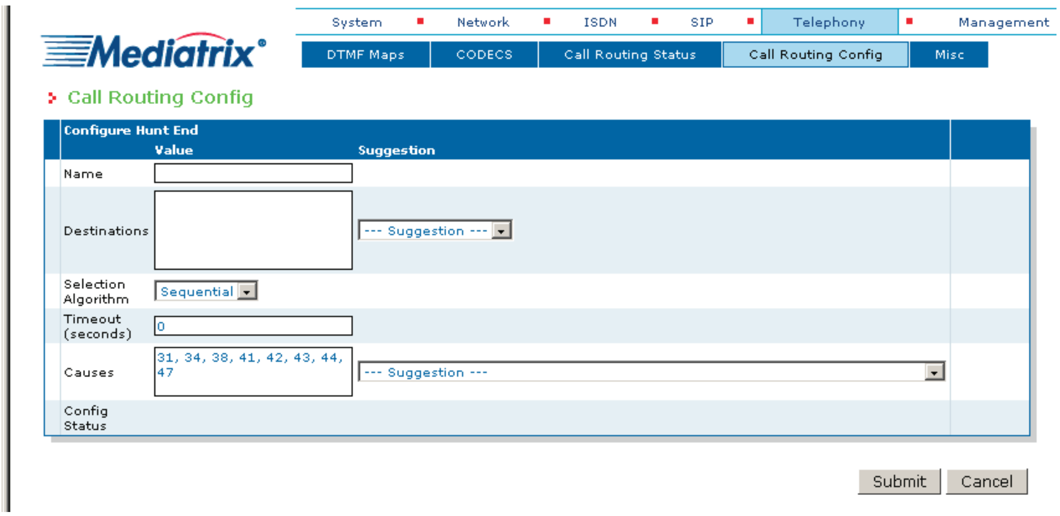
In the current scenario, you will use a Hunt Group to group both of the Mediatrix 4402's ISDN BRI interfaces as one virtual call destination.

1. Click the **Telephony** menu, then the **Call Routing Config** sub-menu. The following window appears.



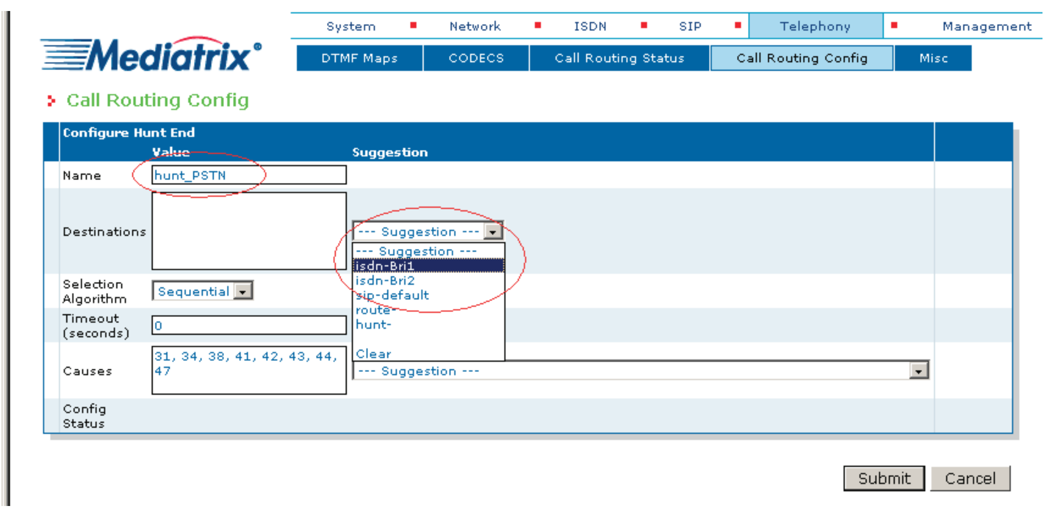
2. Locate the *Hunt* section at the bottom of the window.

3. Click the + button at the bottom right of the *Hunt* section. The following window appears.



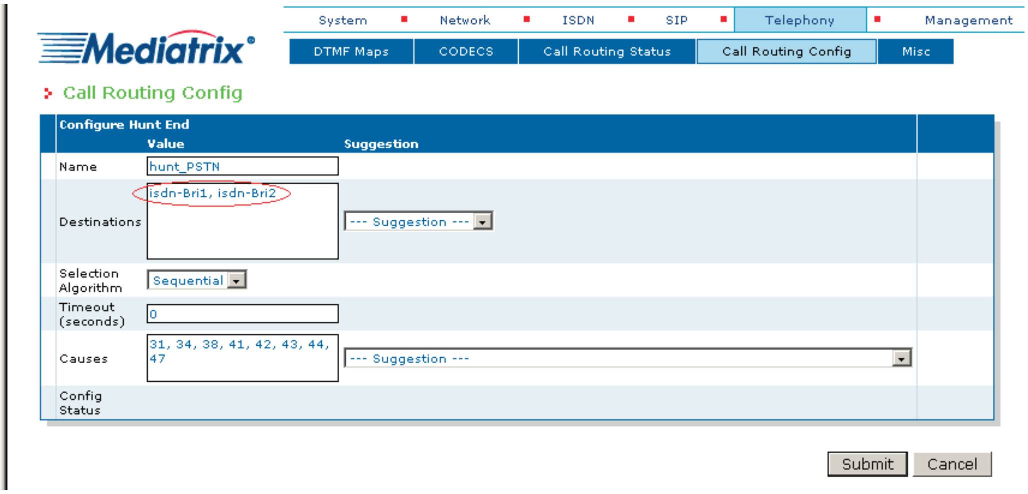
To create a Hunt Group:

4. Set the *Name* field to ***hunt_PSTN***.
5. Use the *Suggestion* drop-down list to select and add the possible destinations that will be part of the Hunt Group.

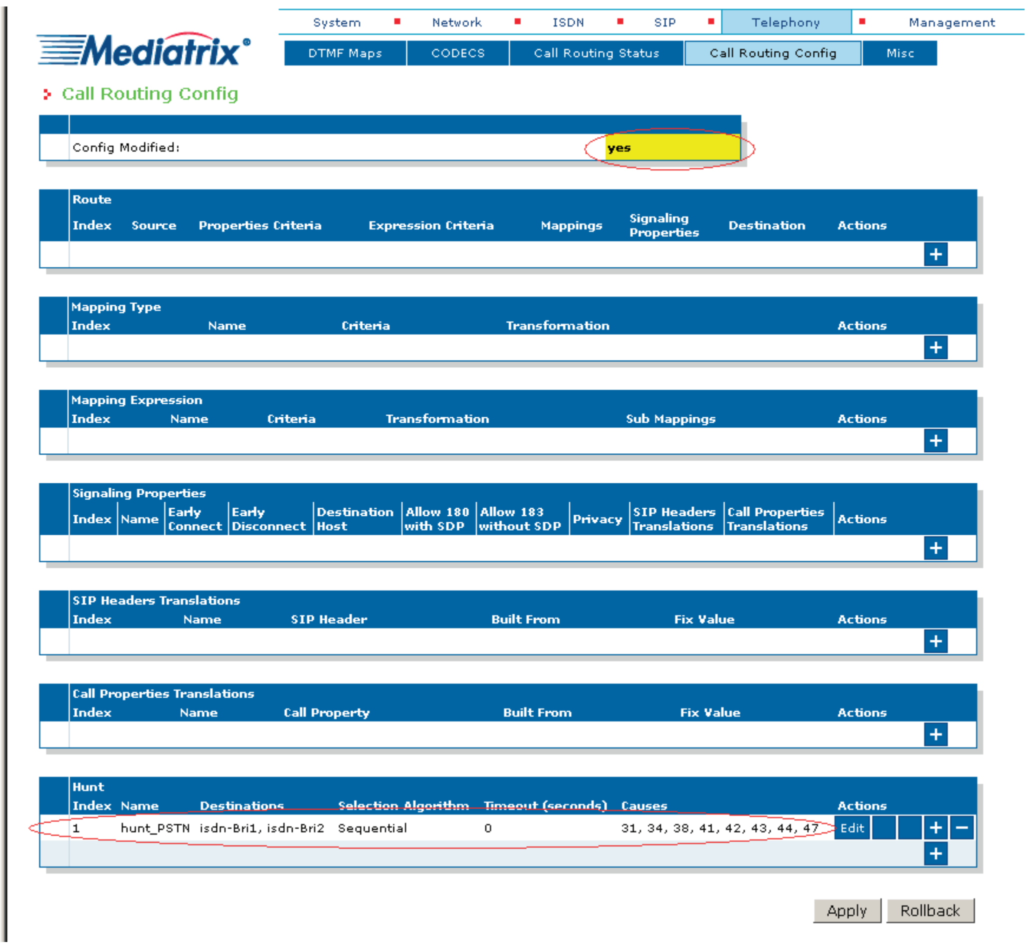


6. Following the Deployment Scenario, select both the Mediatrix 4400's ISDN BRI interfaces (**isdn-Bri1**, which corresponds to port BR1 and **isdn-Bri2**, which corresponds to port BR2) one at a time. The interfaces will be automatically added as destinations for that Hunt Group.

7. Leave the other fields with their default value.



8. Verify whether or not the ISDN interfaces have been successfully added to the configuration by checking the *Destinations* field, then click **Submit** to apply changes and save the new Hunt Group.



9. Your are brought back to the **Call Routing Config** sub-menu. You can see the *Hunt Group* you have just created in the *Hunt* section.

   You can also see a yellow *Yes* warning you that the configuration has been modified but not applied (i.e., the **Call Routing Status** differs from the **Call Routing Config**). The *Call Routing Config* sub-menu is a working area where you build up a Call Router configuration. While you work in this area, the configured parameters are saved but not applied (i.e., they are not used to process incoming calls). The yellow *Yes* flag warns you that the configuration has been modified but is not applied. You will apply the configuration later when it is complete.

Mappings

The purpose of this subsection is to configure Mappings in the Mediatrix 4400, so they can be referred to later when creating routes that need to add/remove a prefix from called phone numbers.

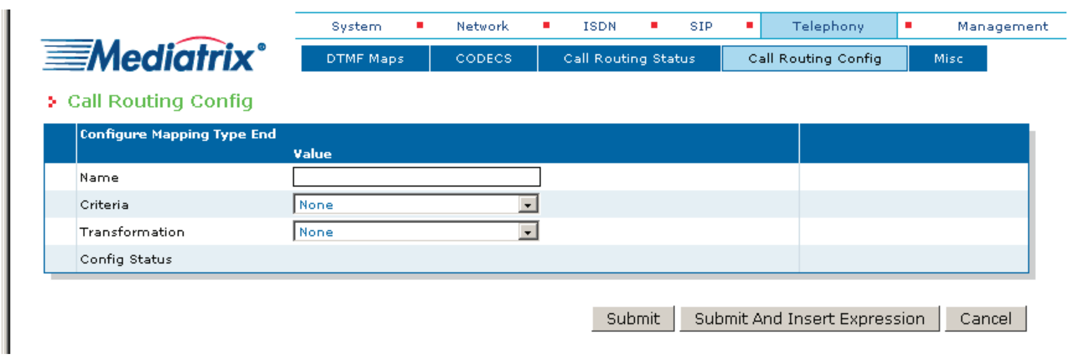
In the current scenario, when calls are routed from the PSTN to SIP, the prefix ***030*** (area code) needs to be removed from the called number. As well, when calls are routed from SIP to the PSTN, the prefix ***030*** needs to be added to the called phone number. You will use Mappings to perform these transformations on the called phone number.

The Mappings are described by two entities in the Mediatrix 4400: Mapping Types and Mapping Expressions. This allows to create virtual Mapping “tables”, where a Mapping Type defines the criteria a call must fulfill to enter the table, and the associated Mapping Expressions describe specific transformation rows in the table.

The most basic Mapping table consists in one Mapping Type and one associated Mapping Expression. In the current scenario, you will create two basic Mapping tables.

1. Locate the *Mapping Type* section.

2. Click the + button at the bottom right of the *Mapping Type* section. The following window appears.

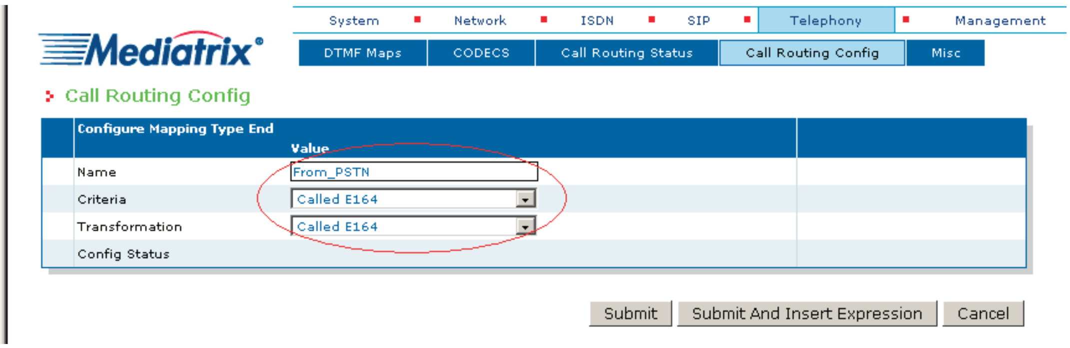


   To create a Mapping Type that applies to incoming calls from the PSTN and alters the called E.164 phone number, proceed as follows:

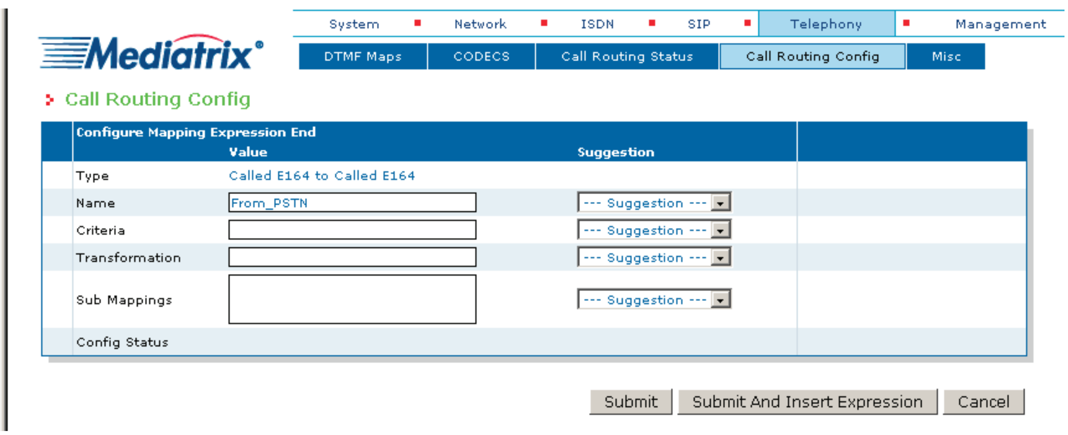
3. Set the *Name* field to ***From_PSTN***.

4. Set the *Criteria* field to **Called E164** using the drop-down list.

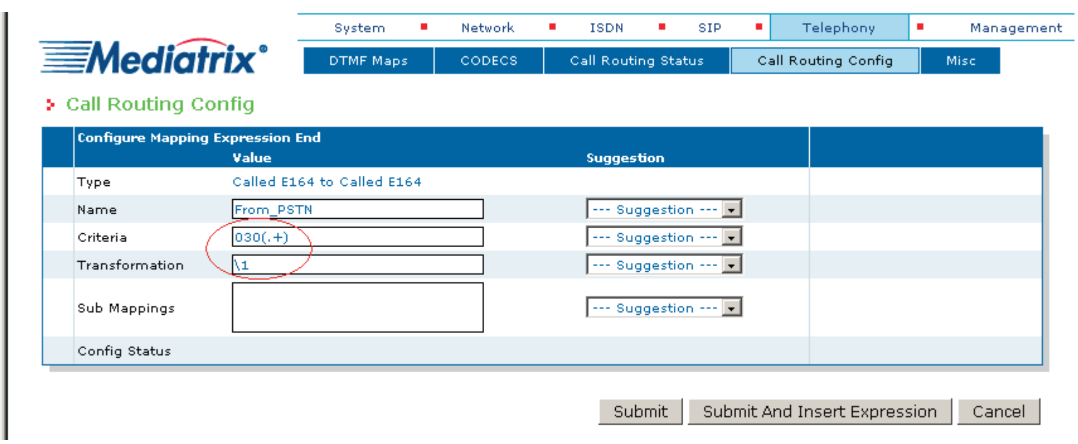
5. Set the *Transformation* field to **Called E164** using the drop-down list.



6. Click **Submit And Insert Expression** to apply changes, save the new Mapping Type, and proceed to enter new Mapping Expressions associated with the Mapping Type just created. The following window appears.



7. To create a Mapping Expression that removes the prefix ***030*** from dialed phone numbers that start with ***030***, set the *Criteria* field to ***030(.+)*** and the *Transformation* field to ***\1***.



This mapping expression applies its transformation to any called E.164 phone number starting with ***030***. The transformation consists in removing the ***030*** prefix from the called E.164 phone number. If the phone numbers in your specific scenario differ, you can modify the contents of the *Criteria* and *Transformation* fields to suit your needs. These fields use the regular expressions syntax (see the section Further Information and Configuration for more details).

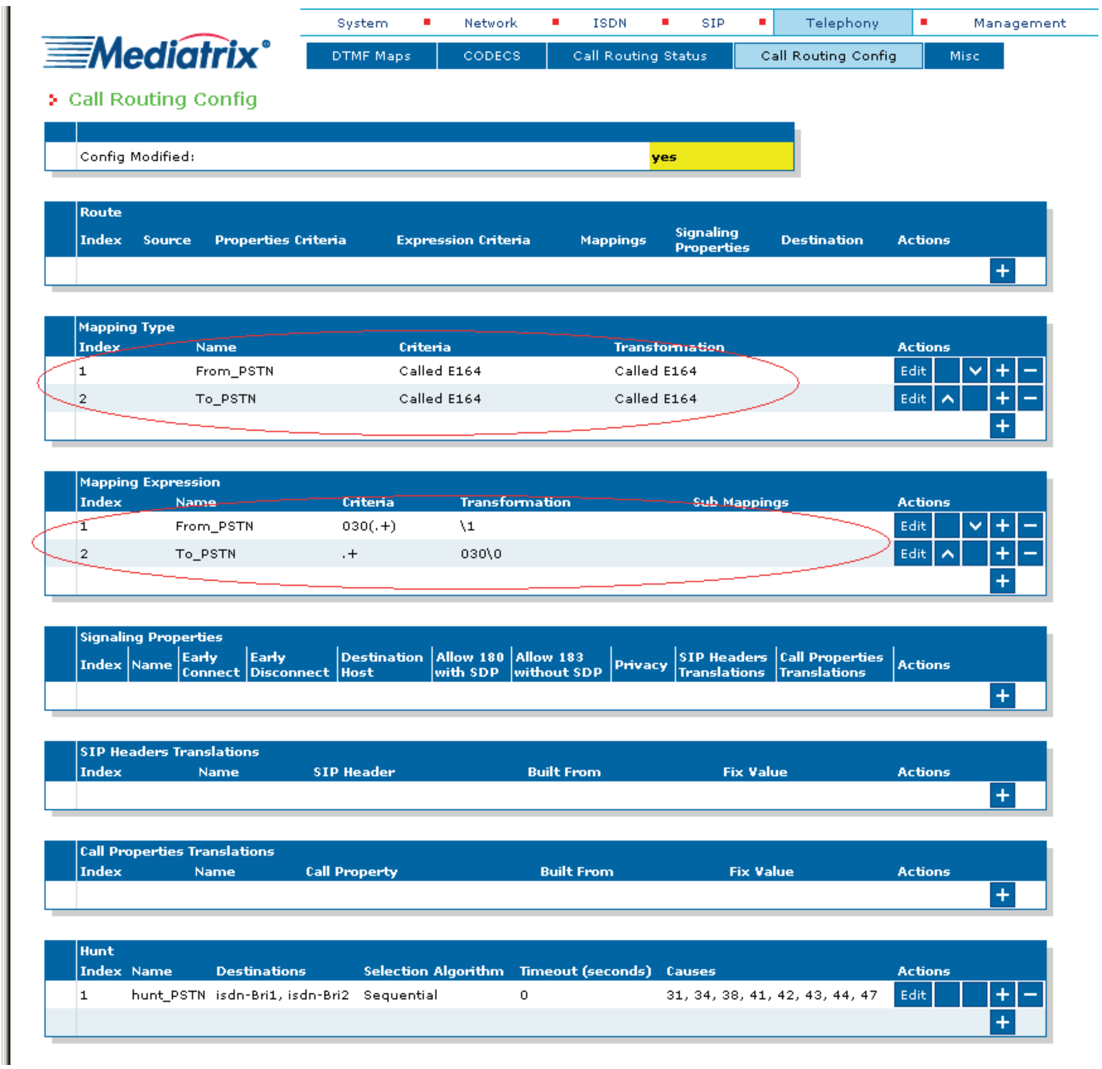
8. Click **Submit** to apply changes and save the new Mapping Expression.

9. Repeat steps 2 to 8 to create an additional Mapping Type and Mapping Expression. These will apply to outgoing calls to the PSTN and add the prefix ***030*** to dialed phone numbers. Use the following field values.

   a. Mapping Type: set the *Name* field to ***To_PSTN***, the *Criteria* field to **Called E164** and the *Transformation* field to **Called E164**.

   b. Mapping Expression: set the *Criteria* field to **.+** and the *Transformation* field to ***030\0***.

10. After completing all the Mapping creation steps, you will see your two Mapping Types and two Mapping Expressions. You will also see the yellow *Yes* warning you that the configuration has been modified but not applied (i.e., the **Call Routing Status** differs from the **Call Routing Config**).

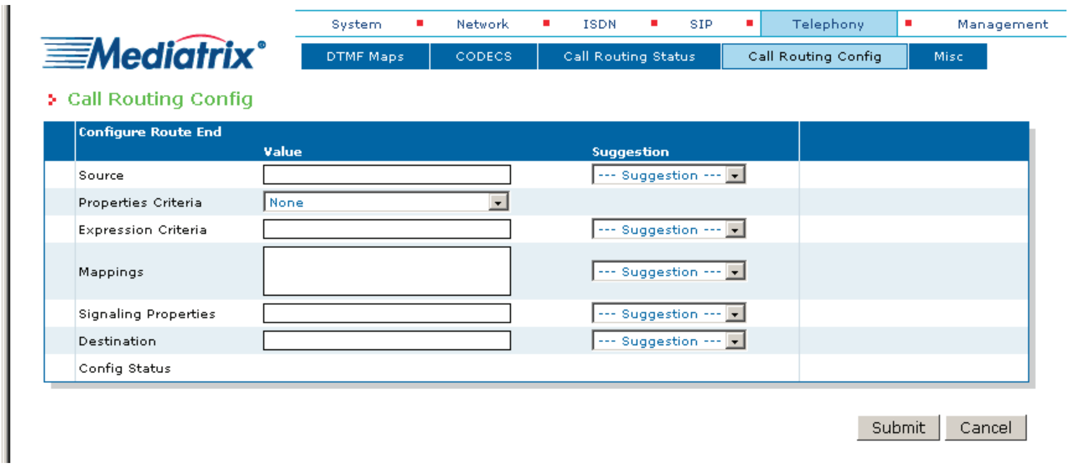


Route

The purpose of this subsection is to configure the Mediatrix 4400 so it makes virtual "connections" between call sources and destinations, using existing Mappings and Hunt Groups as needed.

1. Locate the *Route* section at the top of the window.

2. Click the + button at the bottom right of the *Route* section. The following window appears.



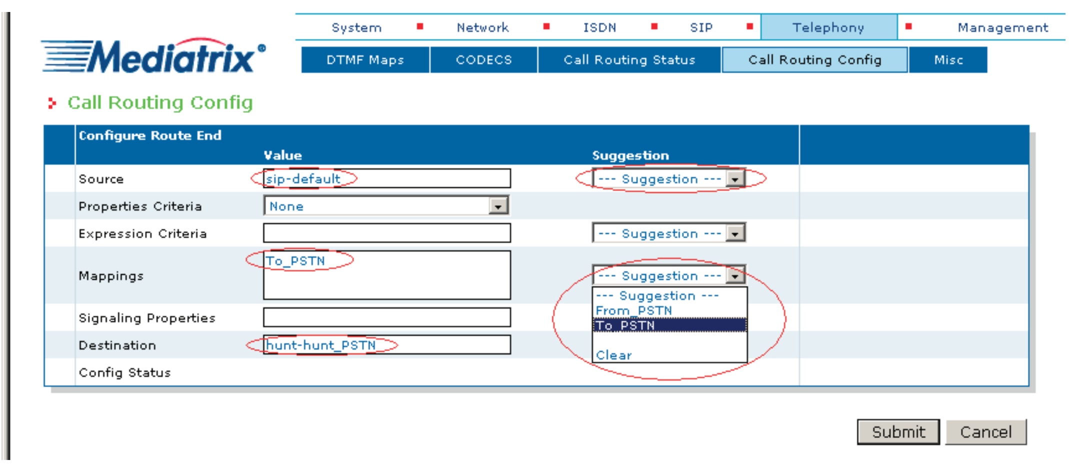
3. To create a route from SIP (**sip-default**) to the PSTN (***hunt_PSTN***):

   o set the *Source* field to **sip-default**.

   o set the *Destination* field to ***hunt-hunt_PSTN***.

   o add the mapping ***To_PSTN*** to the *Mappings* field.

You can use the three fields' associated *Suggestion* drop-down list to help you fill them. This route will satisfy call scenario B described in section Deployment Scenario, where phone 1 calls phone 2.



4. Click **Submit** to apply changes and save the new route.

Config Modified:	yes

Route Index	Source	Properties Criteria	Expression Criteria	Mappings	Signaling Properties	Destination	Actions
1	sip-default	None		To_PSTN		hunt-hunt_PSTN	Edit + −

Mapping Type Index	Name	Criteria	Transformation	Actions
1	From_PSTN	Called E164	Called E164	Edit + −
2	To_PSTN	Called E164	Called E164	Edit + −

Mapping Expression Index	Name	Criteria	Transformation	Sub Mappings	Actions
1	From_PSTN	030(.+)	\1		Edit + −
2	To_PSTN	.+	030\0		Edit + −

Signaling Properties Index	Name	Early Connect	Early Disconnect	Destination Host	Allow 180 with SDP	Allow 183 without SDP	Privacy	SIP Headers Translations	Call Properties Translations	Actions

SIP Headers Translations Index	Name	SIP Header	Built From	Fix Value	Actions

Call Properties Translations Index	Name	Call Property	Built From	Fix Value	Actions

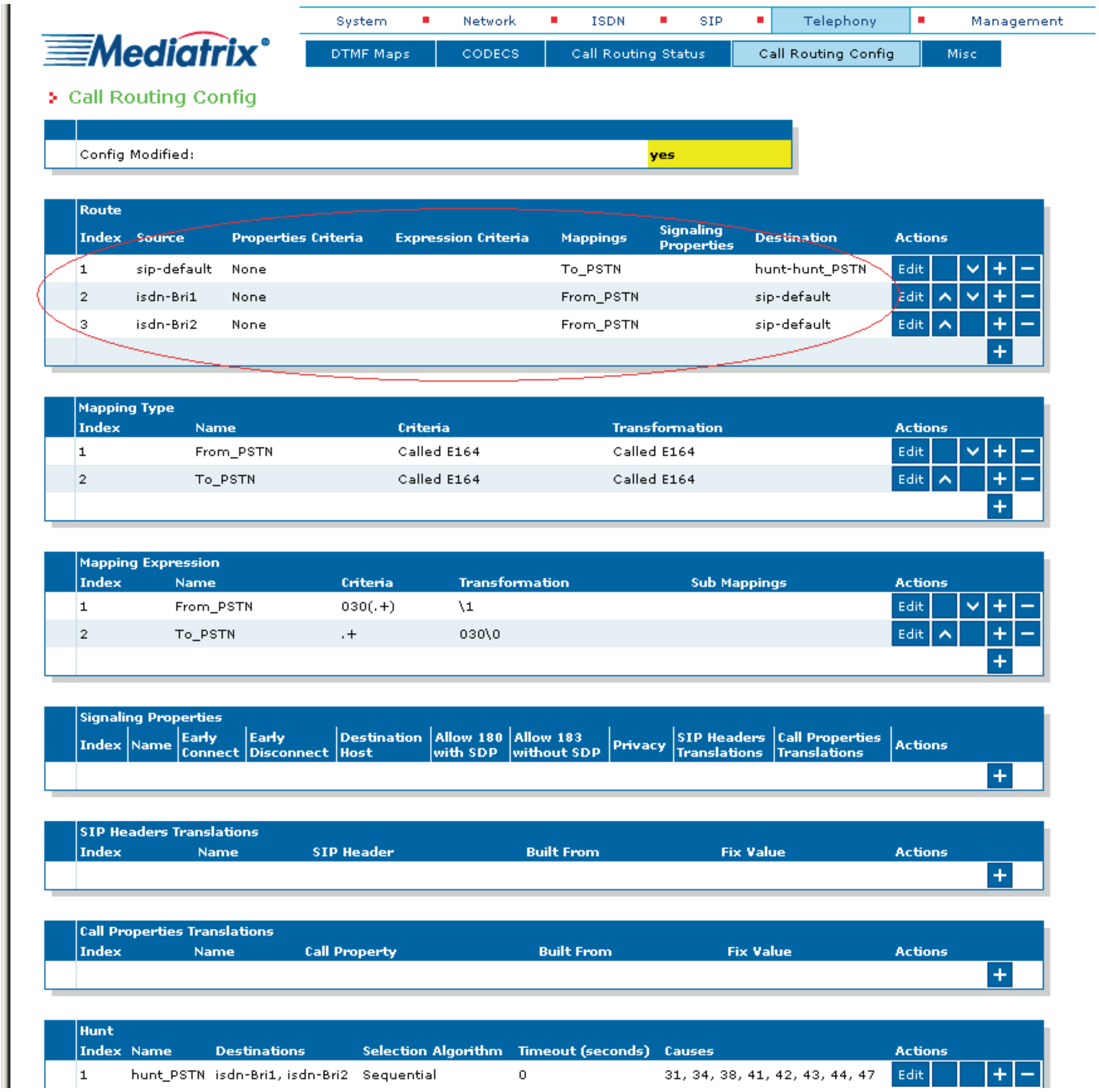
Hunt Index	Name	Destinations	Selection Algorithm	Timeout (seconds)	Causes	Actions
1	hunt_PSTN	isdn-Bri1, isdn-Bri2	Sequential	0	31, 34, 38, 41, 42, 43, 44, 47	Edit + −

5. You are brought back to the **Call Routing Config** sub-menu. You can see the route you just created in the *Route* section. You can also see the yellow *Yes* warning you that the configuration has been modified but not applied (i.e., the **Call Routing Status** differs from the **Call Routing Config**).

6. Repeat steps 2 to 5 twice to create two additional routes. These new routes will satisfy call scenario A described in Deployment Scenario, where phone 2 calls phone 1:

   - one from *Source* **isdn-Bri1** to *Destination* **sip-default** using *Mapping* ***From_PSTN***, and
   - one from *Source* **isdn-Bri2** to *Destination* **sip-default** using *Mapping* ***From_PSTN***.

7. After completing all the route configuration steps, you will see your three routes.



System | Network | ISDN | SIP | Telephony | Management

DTMF Maps | CODECS | Call Routing Status | Call Routing Config | Misc

Call Routing Config

Config Modified:	yes

Route

Index	Source	Properties Criteria	Expression Criteria	Mappings	Signaling Properties	Destination	Actions
1	sip-default	None		To_PSTN		hunt-hunt_PSTN	Edit
2	isdn-Bri1	None		From_PSTN		sip-default	Edit
3	isdn-Bri2	None		From_PSTN		sip-default	Edit

Mapping Type

Index	Name	Criteria	Transformation	Actions
1	From_PSTN	Called E164	Called E164	Edit
2	To_PSTN	Called E164	Called E164	Edit

Mapping Expression

Index	Name	Criteria	Transformation	Sub Mappings	Actions
1	From_PSTN	030(.+)	\1		Edit
2	To_PSTN	.+	030\0		Edit

Signaling Properties

Index	Name	Early Connect	Early Disconnect	Destination Host	Allow 180 with SDP	Allow 183 without SDP	Privacy	SIP Headers Translations	Call Properties Translations	Actions

SIP Headers Translations

Index	Name	SIP Header	Built From	Fix Value	Actions

Call Properties Translations

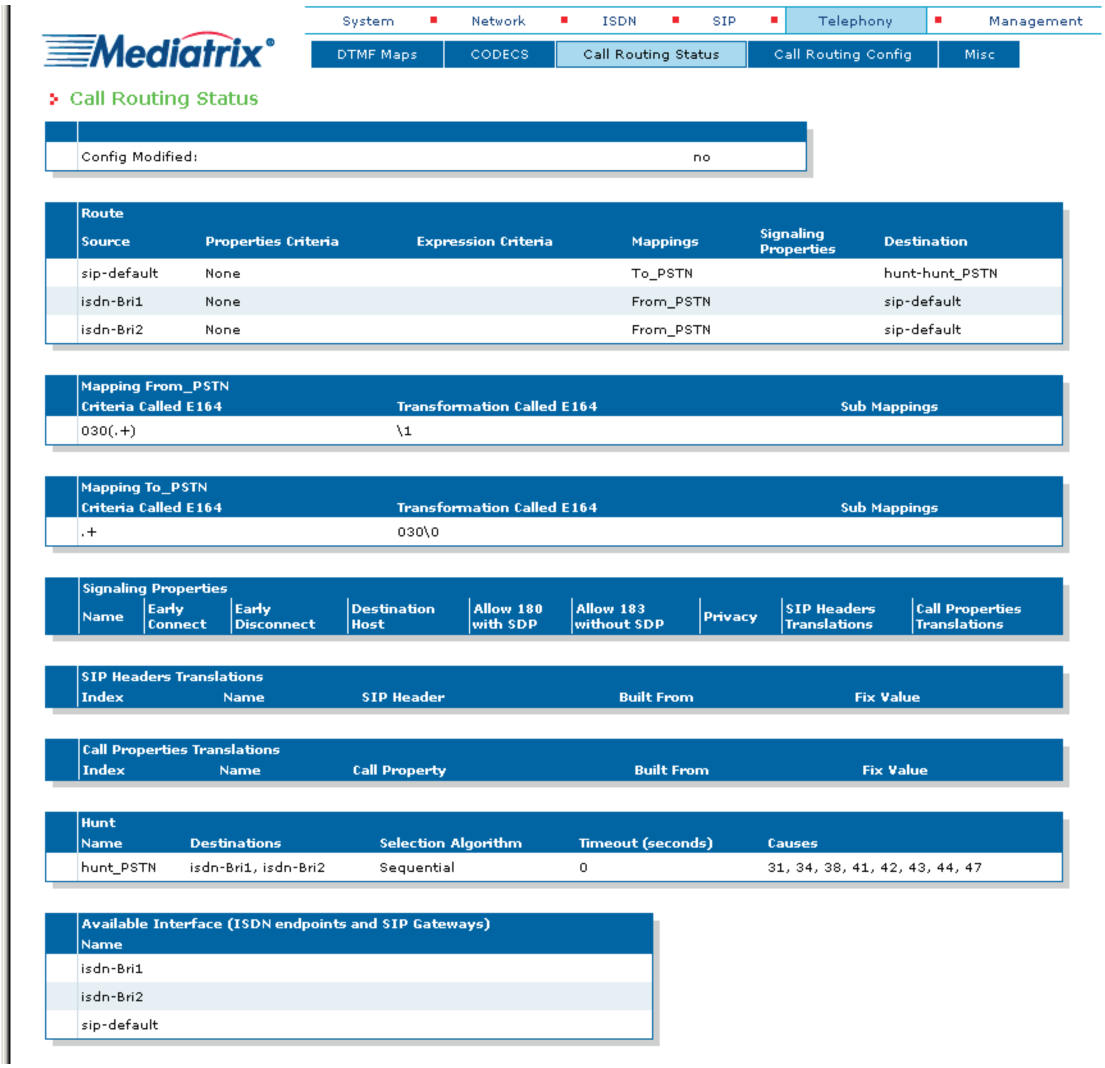
Index	Name	Call Property	Built From	Fix Value	Actions

Hunt

Index	Name	Destinations	Selection Algorithm	Timeout (seconds)	Causes	Actions
1	hunt_PSTN	isdn-Bri1, isdn-Bri2	Sequential	0	31, 34, 38, 41, 42, 43, 44, 47	Edit

8. Click **Apply**. This applies all the parameters from **Call Routing Config** to the system. You can also see that the yellow *Config Modified* **yes** flag is cleared.

9. The call routing parameters can be seen in the **Call Routing Status** window.



System | Network | ISDN | SIP | Telephony | Management

DTMF Maps | CODECS | Call Routing Status | Call Routing Config | Misc

Call Routing Status

Config Modified:	no

Route					
Source	**Properties Criteria**	**Expression Criteria**	**Mappings**	**Signaling Properties**	**Destination**
sip-default	None		To_PSTN		hunt-hunt_PSTN
isdn-Bri1	None		From_PSTN		sip-default
isdn-Bri2	None		From_PSTN		sip-default

Mapping From_PSTN		
Criteria Called E164	**Transformation Called E164**	**Sub Mappings**
030(.+)	\1	

Mapping To_PSTN		
Criteria Called E164	**Transformation Called E164**	**Sub Mappings**
.+	030\0	

Signaling Properties								
Name	**Early Connect**	**Early Disconnect**	**Destination Host**	**Allow 180 with SDP**	**Allow 183 without SDP**	**Privacy**	**SIP Headers Translations**	**Call Properties Translations**

SIP Headers Translations				
Index	**Name**	**SIP Header**	**Built From**	**Fix Value**

Call Properties Translations				
Index	**Name**	**Call Property**	**Built From**	**Fix Value**

Hunt				
Name	**Destinations**	**Selection Algorithm**	**Timeout (seconds)**	**Causes**
hunt_PSTN	isdn-Bri1, isdn-Bri2	Sequential	0	31, 34, 38, 41, 42, 43, 44, 47

Available Interface (ISDN endpoints and SIP Gateways)
Name
isdn-Bri1
isdn-Bri2
sip-default

The configuration note has prepared the system to perform calls in both directions.

Basic Call Establishment

Once this configuration procedure is completed, you are ready to start making basic calls through your new Mediatrix 4400, considering that the rest of your network's setup is configured properly.

Perform Basic Call (Scenario A)

- Pickup phone number 2.
- Dial ***030 695 550***.
- Phone number 1 rings.
- Pick up phone number 1.
- The call is established.
- Hang up both phones to terminate the call.

Perform Basic Call (Scenario B)

- Pickup phone number 1.
- Dial ***695 110***.
- Phone number 2 rings.
- Pick up phone number 2.
- The call is established.
- Hang up both phones to terminate the call.

Further Information and Configuration

You can refer to the following documents/sections for further information on configuration parameters and features used in this configuration note.

All documents are available online on the Mediatrix Download Portal at https://support.mediatrix.com/DownloadPlus/Download.asp.

1- For more information on the Partial Reset feature, and on what to do after performing a Partial Reset to recover a unit with which you have lost contact, refer to the *Partial Reset* section of the *Mediatrix 4400 Digital Gateway Software Configuration Guide*.

2- For more information on configuring level 2 network links, level 3 network interfaces and IP addresses, refer to the *Interfaces Configuration* section of the *Mediatrix 4400 Digital Gateway Software Configuration Guide*.

3- For more information on configuring the Mediatrix 4400's ISDN BRI interfaces in TE or NT mode and additional parameters such as ISDN power feeding, refer to the *ISDN Configuration* section of the *Mediatrix 4400 Digital Gateway Software Configuration Guide*.

4- For more information on configuring the Mediatrix 4400 to work with SIP servers that require SIP authentication, refer to the *SIP Authentication* section of the *Mediatrix 4400 Digital Gateway Software Configuration Guide*.

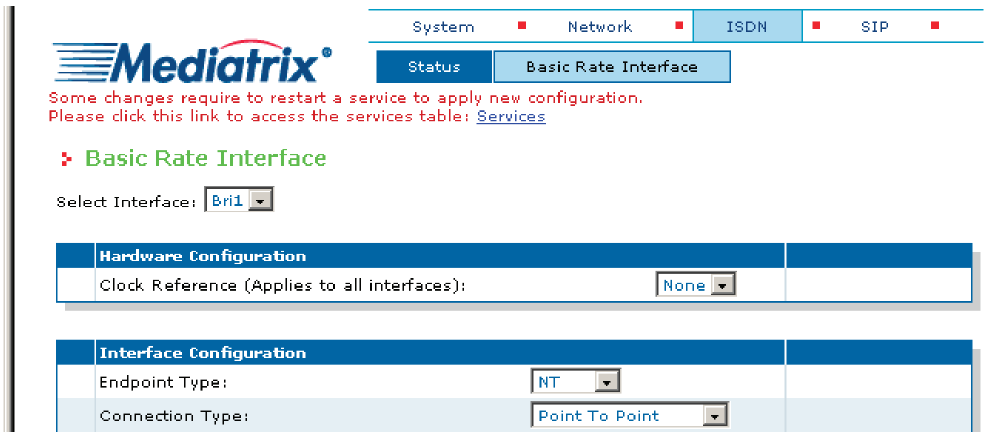
5- For information on how to configure the Mediatrix 4400 so it processes dialed DTMFs according to specific dialing plans, refer to the *DTMF Maps Configuration* section of the *Mediatrix 4400 Digital Gateway Software Configuration Guide*.

6- For more information on call routing including routes, criteria, mappings, signaling properties, hunts, and regular expressions, refer to the *Call Router Configuration* section of the *Mediatrix 4400 Digital Gateway Software Configuration Guide*.

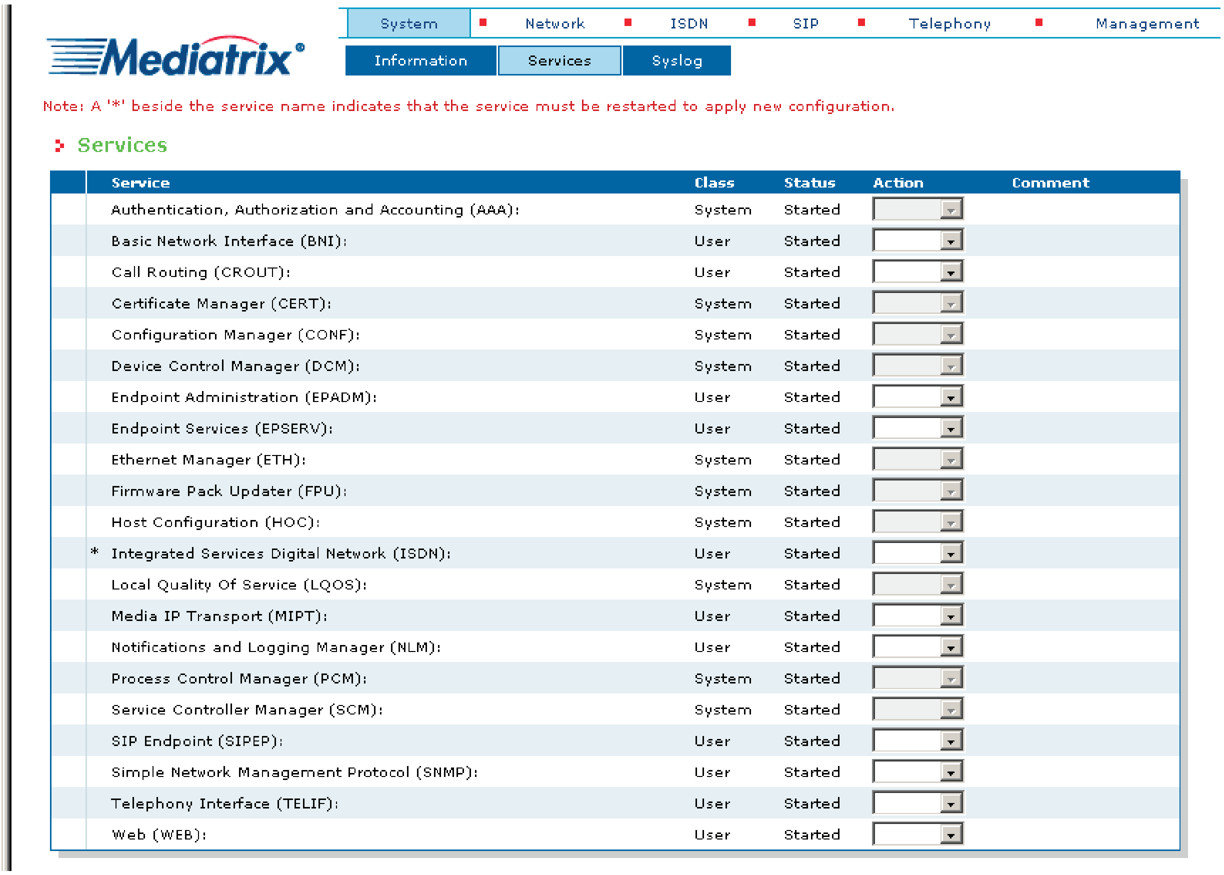
Appendix A - Restarting a Service

The Mediatrix 4400's features are divided in logical entities called *Services.* Some parameters in the Mediatrix 4400 require that the service to which they belong be restarted when they are configured in order for their new configuration value to be correctly applied. When this happens (usually after you click a **Submit** button), a message and a **Services** link are displayed at the top of the window stating that a service must be restarted.

In this example, a parameter of the ISDN services requires that this service be restarted.

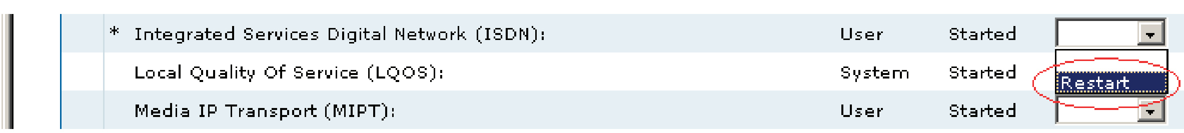


1. Click the **Services** link, which brings you to the **Services** page. In this page, each service that requires to be restarted has a "*" besides its name, as illustrated in the following window.



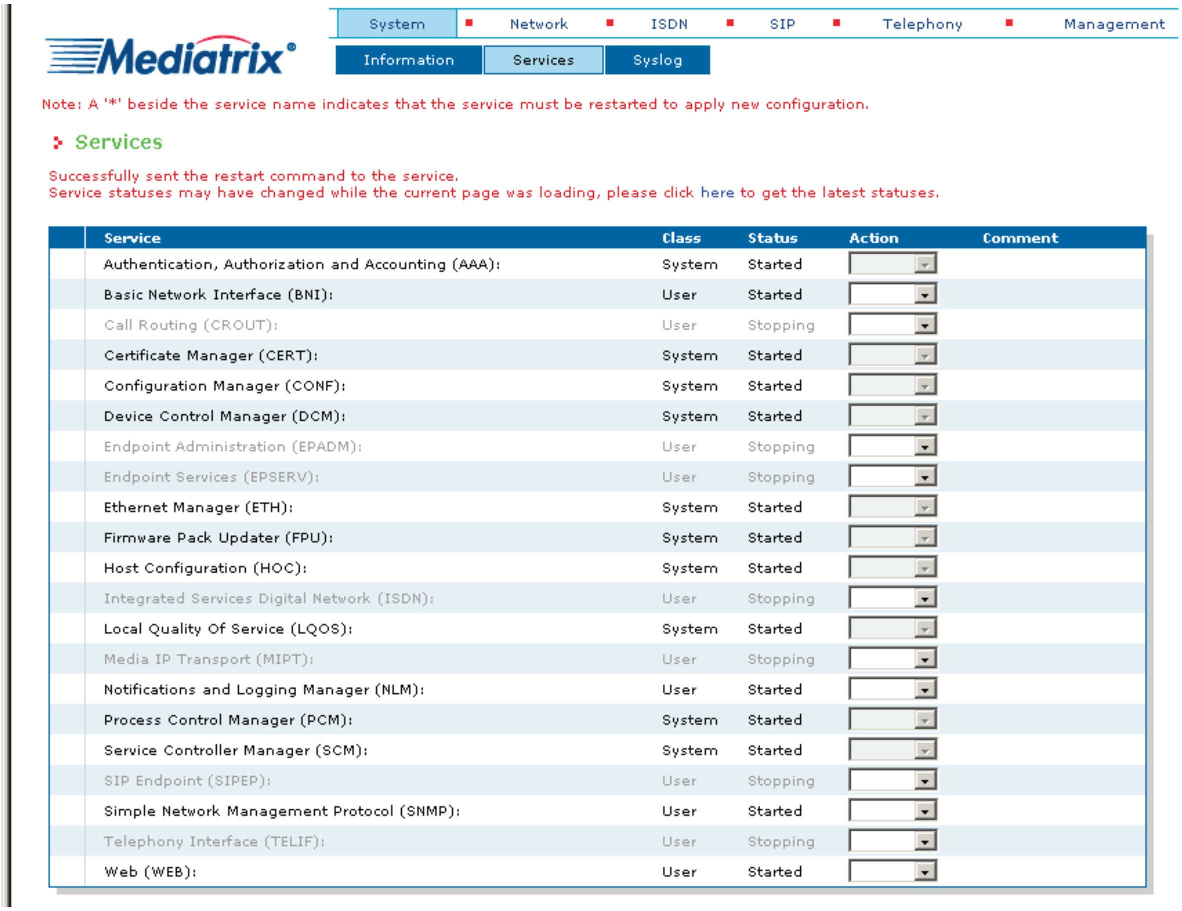
	Service	Class	Status	Action	Comment
	Authentication, Authorization and Accounting (AAA):	System	Started		
	Basic Network Interface (BNI):	User	Started		
	Call Routing (CROUT):	User	Started		
	Certificate Manager (CERT):	System	Started		
	Configuration Manager (CONF):	System	Started		
	Device Control Manager (DCM):	System	Started		
	Endpoint Administration (EPADM):	User	Started		
	Endpoint Services (EPSERV):	User	Started		
	Ethernet Manager (ETH):	System	Started		
	Firmware Pack Updater (FPU):	System	Started		
	Host Configuration (HOC):	System	Started		
*	Integrated Services Digital Network (ISDN):	User	Started		
	Local Quality Of Service (LQOS):	System	Started		
	Media IP Transport (MIPT):	User	Started		
	Notifications and Logging Manager (NLM):	User	Started		
	Process Control Manager (PCM):	System	Started		
	Service Controller Manager (SCM):	System	Started		
	SIP Endpoint (SIPEP):	User	Started		
	Simple Network Management Protocol (SNMP):	User	Started		
	Telephony Interface (TELIF):	User	Started		
	Web (WEB):	User	Started		

2. Restart each service that has a "*" besides its name by clicking the **Restart** action so it correctly applies its new configuration.



*	Integrated Services Digital Network (ISDN):	User	Started	
	Local Quality Of Service (LQOS):	System	Started	Restart
	Media IP Transport (MIPT):	User	Started	

3. Restarting a service may require other services to be restarted. This is why you would see a few services go from the stopping to starting to started states, even if you only restarted one service. The displayed status may be refreshed at any time by clicking the **Services** submenu or the **here** link.



Service	Class	Status	Action	Comment
Authentication, Authorization and Accounting (AAA):	System	Started		
Basic Network Interface (BNI):	User	Started		
Call Routing (CROUT):	User	Stopping		
Certificate Manager (CERT):	System	Started		
Configuration Manager (CONF):	System	Started		
Device Control Manager (DCM):	System	Started		
Endpoint Administration (EPADM):	User	Stopping		
Endpoint Services (EPSERV):	User	Stopping		
Ethernet Manager (ETH):	System	Started		
Firmware Pack Updater (FPU):	System	Started		
Host Configuration (HOC):	System	Started		
Integrated Services Digital Network (ISDN):	User	Stopping		
Local Quality Of Service (LQOS):	System	Started		
Media IP Transport (MIPT):	User	Stopping		
Notifications and Logging Manager (NLM):	User	Started		
Process Control Manager (PCM):	System	Started		
Service Controller Manager (SCM):	System	Started		
SIP Endpoint (SIPEP):	User	Stopping		
Simple Network Management Protocol (SNMP):	User	Started		
Telephony Interface (TELIF):	User	Stopping		
Web (WEB):	User	Started		

Thank you for using Mediatrix solutions!